SIGNAL PROCESSING IN A HEARING AID

The invention relates to a device and a method for the signal processing in a hearing aid in accordance with the preamble of the independent claims The invention is suitable in particular for the improvement of the language comprehensibility by the suppression of interfering noise in the case of hearing aids, resp., hearing devices.

STATE OF THE ART

A method in accordance with the field of the invention is known, for example, from EP 1 067 821 A1, the contents of which are herewith incorporated into this application. In it an acoustic aid is described, in which the suppression of interfering noise takes place in a main signal path, which comprises neither a transformation in the frequency range nor a splitting-up into partial band signals, but solely comprises a suppression filter. A transmission function of the suppression filter is periodically determined anew on the basis of attenuation factors, which are established in a signal analysis path, which lies parallel to the main signal path. The attenuation factors are utilised for the attenuation of signal components in frequency bands having a significant proportion of interfering noise. The suppression filter is implemented as a transverse filter, the pulse response of which is periodically calculated anew as the weighted sum of the pulse responses of transverse band pass filters. In this manner, a processing with little signal delay is possible.

DESCRIPTION OF THE INVENTION

It is an object of the invention to create a device and a method for the signal processing in a hearing aid of the kind mentioned above, which implement a higher quality and comprehensibility of the processed signal.

This object is achieved by a device and a method for the signal processing in a hearing aid with the features of the claims 1 and 10 as well as a hearing aid with the features of the claim 20.

In the method according to the invention for the signal processing in a hearing aid

- coefficients of a compression amplification, which describe a frequency-dependent adaptation of the input signal in accordance with frequency-dependent signal levels of the input signal, are determined,
- coefficients of a noise suppression, which describe a frequency-dependent adaptation of the input signal in accordance with interfering noise detected in the input signal, are determined, and
- coefficients of a filter for the filtering of the input signal are calculated from the coefficients of the compression amplification and the coefficients of the noise suppression.

In this, with the term "adaptation of a signal" in summary both an amplification as well as an attenuation are meant.

By means of the invention it becomes possible to adapt the amplitude characteristic of the filter to changing voice signals and interference signals as well as to the

requirements of a person with poor hearing, wherein a delay time for the filtering of the input signal is kept short.

A further advantage is that the compression amplification allows differing amplification values for different frequency ranges of the input signal.

A further advantage is the fact that only a single controllable filter is utilised both for the compression amplification as well as for the noise suppression.

In a preferred embodiment of the invention, determining the coefficients of the compression amplification takes place in a first number of frequency ranges Fn with n=1..N of the input signal on the basis of signal levels or amplitude components. A signal level is determined from a partial signal of the input signal, which is formed by filtering the input signal and splitting it up into partial signals with signal components respectively in only one frequency range. The signal levels are iteratively determined as momentary effective values of a signal power in the respective frequency ranges of the input signal. As a result, it becomes possible to adapt the compression amplification with a time-dependent resolution that corresponds to a sampling rate of the input signal.

In a preferred embodiment of the invention determining the coefficients a_m of the noise suppression takes place in a second number of frequency ranges Φ_m with m=1..M of the input signal by determining modulation depths d_m and by determining the coefficients a_m for each one of the frequency ranges Φ_m in accordance with the corresponding modulation depth d_m. In doing so, the modulation depths d_m are determined from a time-dependent sequence of maximum – and minimum values of a signal level p_m in the corresponding frequency range Φ_m. As a result, it becomes possible to selectively filter out weakly modulated, this means monotonous interfering noises. Time constants for the adaptation of the noise suppression are preferably situated in the range of around 50 milliseconds or below.

In a preferred embodiment of the invention, the frequency ranges Φ_m for the noise suppression are small in comparison with the frequency ranges F_n for the compression amplification. Therefore at least one frequency range F_n comprises two or more frequency ranges Φ_m. Correspondingly, filters for determining proportions of the input in the frequency ranges Φ_m comprise a greater signal run time or delay time than filters for the frequency ranges F_n. This makes possible a distinct split-up of the frequency range for the suppression of interferences and simultaneously a rapid adaptation of the compression amplification to a changing voice signal. A maximum delay which may be tolerated for the adaptation of coefficients of the compression amplification amounts to 5 milliseconds, preferable are values below 2.5 milliseconds. In accordance with the invention, values of below one millisecond are capable of being achieved.

In a further preferred embodiment of the invention, the filter is not exactly updated to the newly calculated coefficients in every sampling interval. Instead of this, it is only updated in accordance with one or several changed coefficients. This enables an adaptation with a small calculation effort and a correspondingly reduced energy consumption. Preferably the adaptation only takes place for that coefficient or those coefficients, the change of which exceed a predefined threshold or which is comparatively great or, respectively, the greatest. Also possible is a periodical changing of respectively one or of some few coefficients or a pseudo-random running through and adaptation of all coefficients.

In a further preferred embodiment of the invention, an influence of the noise suppression is taken into consideration in determining the coefficients for the compression amplification. For this purpose, a means for determining coefficients of the noise suppression transmits correction values to a means for determining coefficients of the compression amplification, which correction values correspond to a signal attenuation caused by the noise suppression.

The device according to the invention comprises the features of claim 10. A hearing aid in accordance with the invention comprises means for the implementation of the method according to the invention.

Further preferred embodiments follow from the dependent claims. In this, characteristics of the method claims are combinable analogously with the device claims and vice versa.

BRIEF DESCRIPTION OF THE DRAWINGS

In the following, the object of the invention is explained in more detail on the basis of preferred examples of embodiments, which are illustrated in the attached drawings. These depict:

Figure 1	schematically a structure of the signal processing;
Figure 2	a block diagram of a calculation of amplification values; and
Figure 3	a block diagram of a calculation of attenuation values and correction values in accordance with the invention.

The reference marks and their significance are listed in the list of reference marks in a summary form. In principle, identical components are referred to in the Figures with identical reference marks.

DESCRIPTION OF PREFERRED EMBODIMENTS

Figure 1 schematically illustrates a structure of the signal processing in a hearing aid according to the invention. An input signal X is brought to a controllable filter 6, to a means for the determination of a compression amplification 7 and to a means for the determination of a noise suppression 8. The controllable filter 6 is designed for the formation of an output signal Y in accordance with filter coefficients $c_1..c_M$.

In the means for the determination of the compression amplification 7, the input signal X is brought to a first filter unit 1. The first filter unit 1 is designed for the determination of signal proportions $x_1..x_N$ of the input signal X in a first number of frequency ranges F_n with $n=1..N$. In a signal processing for the compression amplification 3, from the signal proportions $x_1..x_N$ parameters, respectively, coefficients or adaptation values of the compression amplification $g_1..g_M$ are calculated. These coefficients, with a view to the amplification function of the hearing aid, are also designated as *amplification* values. Other coefficients, however, are also designated as amplification values.

In the means for the determination of the noise suppression 8 the input signal X is brought to a second filter unit 2. The second filter unit 2 is designed for the determination of signal proportions $y_1..y_M$ of the input signal X in a second number of frequency ranges Φ_m with $m=1..M$. In a signal processing for the noise suppression 4, from the signal proportions $y_1..y_M$ parameters, respectively coefficients or adaptation values of the noise suppression $a_1..a_M$ are calculated. These coefficients with a view to the noise suppression achieved are also designated as *attenuation values*.

The combination unit 5 combines the coefficients of the compression amplification $g_1..g_M$ with the coefficients of the noise suppression $a_1..a_M$ and from this calculates combined logarithmic amplification values $c_1..c_M$ as filter coefficients of the

controllable filter 6. Preferably, the mentioned coefficients g_i,a_i and c_i are logarithmically scaled and in the combination unit 5 essentially a subtraction c_m=g_m-a_m with m=1..M is carried out.

In a preferred embodiment of the invention the signal processing for the noise suppression 4 transmits correction values $r_1..r_N$ to the compression amplification 3, which correspond to a respective signal attenuation in the frequency ranges $F_1..F_n$ caused by the noise suppression.

In a further preferred embodiment of the invention, the first filter unit 1 and the second filter unit 2 are not implemented as separate units, but rather as a combined filter unit. For example, sequentially a filtering with wide frequency bands is carried out for the determination of the signal proportions $x_1..x_N$, and these filtered signals are further filtered for the determination of the signal proportions $y_1..y_M$.

The invention in the demonstrated embodiment in summary operates as follows: The input signal is split-up into three signal paths, a main signal path with a controllable filter, a first parallel signal analysis path for the compression amplification and a second parallel signal analysis path for the noise suppression.

Figure 2 depicts a block diagram of a calculation of amplification values in the signal processing for the compression amplification 3. For the compression amplification, signal levels are calculated in N relatively few frequency ranges. Figure 2 illustrates the calculation for *one* of these N frequency ranges, for the remaining frequency ranges the same structure is utilised. From a signal proportion x_n in this frequency range a signal power is formed in a block 21, for example, as a running total of squared signal values. In a block 22, by means of taking the logarithm, a signal level p_n is formed. The term signal level here therefore designates the effective value of the momentary signal power in the frequency range F_n expressed in a logarithmic range of numbers, e.g., in dB. From the signal level p_n by subtraction 23 of a

correction value r_n a modified signal level p_n' is calculated. The determination of correction values r_n is separately dealt with further below. Assigned to every frequency range F_n of the compression amplification is at least one frequency range Φ_m of the noise suppression. For each one of these assigned frequency ranges Φ_m (in Figure 2 there are three, corresponding to blocks 24, 24', 24") a function f_m of its own is predefined, which calculates from the modified signal level p_n' an amplification value g_m, thus

$$g_m = f_m(p_n') .$$

These functions f_m take into account an individual loss of hearing power and audiological experience. Parameters contained in the functions f_m, amplification values or hearing correction values are preferably user-specific and, for example are stored in an EPROM of the hearing aid. The total number of these functions f_m and of the amplification values g_m, that is, over all N frequency ranges F_n of the compression amplification, is equal to the number M of the frequency ranges Φ_m of the noise suppression.

If one is aiming for amplifying quiet phonemes, i.e., consonants, more than loud phonemes, i.e., vowels, in order that for a person with impaired hearing all phonemes in continuously spoken language become audible to an as great as possible extent, then the signal levels p_n have to be determined in such a manner that differences between quiet and loud successive phonemes are well detected. In addition, the continuously determined amplification values g_m have to be applied with the correct timing to those signal sections in which the accompanying phonemes are situated, i.e., the amplification values have to act on the audio signal X synchronously. A synchronous compression amplification acting with such a speed, in the rhythm of successive phonemes only provides good results, if the number of separate frequency ranges is selected to be small, e.g., $N \leq 5$, preferably $N \leq 3$. Otherwise spectral differences between the frequency ranges characteristic for the different phonemes are diminished too much and with this the speech comprehensibility is impaired. The compression amplification with few, relatively wide frequency bands is possible with

a slight processing delay in the order of magnitude of 1 millisecond, which comes close to the requirement of an ideally delay-free signal processing. In a preferred embodiment of the invention, the compression amplification is carried out for only a single frequency band, that is, jointly for the entire frequency range of the audio signal. In another embodiment of the invention, two frequency bands are utilised for this, therefore N=2.

The signal analysis for the determination of signal levels in frequency ranges f_n for the compression amplification is preferably carried out iteratively, wherein for every new value of the input signal current signal levels are determined. For this purpose, preferably recursive signal analysis methods are utilised. For example, the squared average value of the signal x[k] at the k-ed sampling point in time is calculated iteratively as

$$s[k] = s[k-1] + \varepsilon \cdot (x^2[k] - s[k-1]),$$

wherein $0 < \varepsilon << 1$ is selected.

A corresponding signal level value, e.g., in dB, then results as

$$p[k] = 10*\log10(s[k]) .$$

In case of the noise suppression, the objective is to diminish partial signals in frequency ranges of the audio signal, in which frequency ranges mainly only monotonic interfering noises are located. To do so, first of all in M separate frequency ranges Φ_m differences between maximum – and minimum values of the signal levels p_m succeeding one another in time, so-called modulation depths d_m, are established, wherein $m = 1, \ldots, M$ is applicable.

For the noise suppression, an iterative determination of the signal levels in Step with the sampling rate of the input signal is not necessary. In order to save calculation operations, one therefore preferably works with reduced sampling rates. In doing so, the signal level p_m is formed in the corresponding frequency range Φ_m segmentwise for segments with a length of approx. 20-30 ms as the momentary effective value of

the signal power. With this, it is possible keep the noise suppression updated with a resolution in time p_m of, for example, less than 50 ms.

For the determination of maximum values and minimum values, separate estimated value functions are kept updated: For this purpose, in every scanning interval a stored maximum value is either linearly or in accordance with an exponential function reduced by a small increment, or else the current level value is taken over, providing it exceeds this reduced maximum value. In the same manner the minimum value in every sampling interval is increased by a small increment or else the current level value is taken over, providing it falls below the increased minimum value. The modulation depth therefore results as the difference between these two estimated value values. A small modulation depth therefore is produced in case of a signal energy which remains the same. In order to avoid sudden changes in the modulation depth, the difference values established in this manner are preferably additionally subjected to a smoothing. By means of a corresponding selection of the mentioned increments, the extremes decay with time constants in the range of some few seconds.

For speech in a quiet acoustic environment, the modulation depth assumes values of 30 dB and more. In traffic noise, the low frequency range up to around 500 Hz is frequently dominated by a monotonic interfering noise, so that even in case of the presence of speech signals the modulation depth in this frequency range declines to close to 0 dB. Other interfering noises again cover over the speech signal rather more in higher frequency ranges. Preferably partial signals in frequency ranges Φ_m are diminished, in which the modulation depth d_m drops below a critical value of, e.g., 15 dB, wherein the extent of the attenuation a_m monotonically and, for example, linearly increases with a modulation depth becoming smaller.

For an as accurate as possible recording and separation of frequency ranges with differing modulation depths, a large number of separate frequency ranges is

advantageous, e.g., M = 20. For the signal processing in so many narrow frequency bands perforce a long time delay in the order of magnitude of 10 ms results, which, however is still well compatible with a gradual attenuation and occasional increasing of the partial signals in these frequency ranges.

The amplification values g_m of the compression amplification 3 and the attenuation values a_m of the noise suppression 4 are combined for each frequency range and brought to the controllable filter 6 as control variables c_m in the main signal path. The transmission function of the controllable filter when so required is updated in every sampling interval of the input signal, frequency-specific in one or in a few frequency ranges and left unchanged in all other frequency ranges.

For the combined application of compression amplification and noise suppression there is the possibility to carry out a signal analysis in relatively many frequency ranges $\Phi_{m,}$ as it makes sense for the noise suppression, and to thereafter summarise the results in a suitable manner with respect to the few frequency ranges F_n relevant for the noise suppression. The disadvantage of a sequential procedure of this kind consists of the fact, that for the overall signal processing a long signal delay in the order of magnitude of 10 ms results. From the point of few of the calculation effort, for an implementation of this type in particular the fast Fourier transformation and the inverse fast Fourier transformation would appear to be attractive. In doing so, the audio signal one after the other in individual segments with a duration of approx. 10 ms in the frequency range is transformed, analysed and modified, and subsequently transformed back into the time range. By the application of the segment by segment signal processing, however, the following disadvantages result: The signal levels p_n are calculated as average values in a segment, as a result of which a distinctive signal increase at a certain point in time is only recorded with the time-dependent resolution of a processing segment. Also the determination of the individual amplification values and with this of the overall transmission function only takes place at the cadence of the successive segments.

Therefore, the filtering of the input signal X is preferably carried out on the basis of a separate and running in parallel signal analysis for the noise suppression as well as for the compression amplification. In doing so, the coefficients a_m for the noise suppression, that are perforce received with a time delay, are combined with more rapidly received coefficients for the compression amplification g_m, and several of the coefficients g_m with differing functions f_m are determined on the base of the same, optionally modified signal level $p_n'=p_n-r_n$ of a frequency range F_n for the compression amplification.

The combined and parallel processing takes place in detail as follows: In the lowest signal path the audio signal passes through a controllable filter 6, which carries out the necessary frequency-dependent signal modifications. The two upper signal paths each contain a filter unit, which filter units split-up the audio signal into partial signals of separate frequency ranges. The first filter unit 1 effects a signal split-up in only few frequency ranges F_n with the width N, which can be implemented with an only slight signal delay. The second filter unit 2 effects a signal split-up into many frequency ranges Φ_m with a narrow width M, which entails a long delay time. In doing so, the frequency ranges are preferably selected in such a manner that every frequency range Φ_m is a partial range of a frequency range F_n . The frequency ranges for the compression amplification F_n together preferably cover the same frequency range as the frequency range for the noise suppression Φ_m. a frequency range for the compression amplification respectively covers several frequency ranges for the noise suppression. Ratios between the widths of frequency ranges and between the splitting-up of frequency ranges are preferably at least nearly logarithmic.

A typical frequency range for the input signal is: 0 to 10 kHz. This is, for example, split-up into the following frequency ranges for the compression amplification and the noise suppression:

Compression amplification (Hz)	Noise suppression (Hz)
0 to 1250	0 to 312.5
	312.5 to 625
	625 to 937.5
	937.5 to 1250
1250 to 2500	1250 to 1562.5
	1562.5 to 1875
	1875 to 2187.5
	2187.5 to 2500
2500 to 10000	2500 to 3125
	3125 to 3750
	3750 to 4375
	4375 to 5000
	50000 to 6250
	6250 to 7500
	7500 to 10000

In this, the sampling rate amounts to, for example, 20 kHz and correspondingly the useful band width to half of that, therefore 10 kHz. In another embodiment of the invention, these values amount to 16 kHz, respectively, 8 kHz.

In the signal analysis for the noise suppression, for every one of the M frequency ranges Φ_m a determination of the assigned signal level p_m, of the modulation depth d_m and of the attenuation value a_m takes place, wherein the latter is advantageously expressed in a logarithmic range of numbers. The determination of the modulation depth d_m takes place as described above in accordance with, i.e., as a function of the time-dependent characteristic of the corresponding signal level p_m, and the determination of the coefficients a_m in accordance with the corresponding modulation depths d_m. The second filter unit 2 and a part of the signal processing for

the noise suppression 4 therefore form a means for determining these values p_m, d_m and a_m in a second number of frequency ranges of the input signal X.

In the signal analysis for the compression amplification, in each of the N frequency ranges F_n the signal level p_n is determined and this in such a manner that every signal value of the partial signal $x_n[k]$ contributes to an updating of the signal level, which leads to a higher time-dependent resolution than in the case of the sole determination of a segment by segment average value.

The first filter unit 1 and a part of the signal processing for the compression amplification 3 therefore form a means for the determination of signal levels in a first number of frequency ranges of the input signal X. Subsequently for all M frequency ranges Φ_m amplification values

$$g_m = f_m(p_n')$$

are determined, wherein every modified signal level p_n', thus the levels reduced by the correction values $r_1..r_N$, is utilised for determining the amplification values in all those frequency ranges Φ_m, which in combination result in the frequency range F_n. The correction values r_n take into account a possible reduction of the signal powers as a result of the noise suppression.

Each one of the amplification values g_m with $m = 1..M$ is therefore assigned to a frequency range Φ_m. With the determination of M different amplification values for the narrow frequency ranges Φ_m the compression amplification in the combined signal processing in accordance with the invention is capable of being implemented at the same time also with an essentially more flexible transmission function, therefore with M instead of only N functions f_m, than if solely one amplification value were to be determined for every wide frequency range F_n. The amplification values g_m once again preferably are expressed in a logarithmic scale. The functions f_m determine, frequency-specifically and in dependence of the signal level, a desired frequency-specific amplification in accordance with audiological principles.

The M amplification values and attenuation values reach the combination 5 of amplifications and attenuations, where they are separately combined in every frequency range Φ_m, which in the case of the utilisation of a logarithmic range of numbers takes place by a simple subtraction:

$$c_m = g_m - a_m.$$

The M combined logarithmic amplification values c_m reach the controllable filter 6, where they are transformed into linear amplification values γ_m. The controllable filter 6 with the transmission function H(z) can be assembled out of M parallel filters, the transmission functions $H_m(z)$ of which respectively only in the frequency range Φ_m possess a pass-through characteristic, and in all other frequency ranges have a blocking characteristic, and for the achievement of the desired frequency-dependent modification of the audio signal X are each respectively multiplied with the linear amplification value γ_m

$$H(z) = \gamma 1 \cdot H1(z) + \gamma 2 \cdot H2(z) + \ldots\ldots + \gamma M \cdot HM(z).$$

For an updating of the controllable filter 6 in step with the sampling rate of the audio signal X, this elementary relationship is not suitable, because the calculation effort and the power requirement of an integrated circuit associated with this would be much too great. It is solely suitable for a segment by segment updating, which, however, because of the reduced time-dependent resolution is not optimal in the embodiment illustrated here as an example.

In order to achieve better time-dependent resolution, the transmission function H(z) of the controllable filter 6 preferably is updated iteratively in every sampling interval k in accordance with

$$H(z)[k] = H(z)[k - 1] + \delta H(z)[k],$$

wherein the value $\delta H(z)[k]$ represents the exact updating of the controllable filter 6 in one or perhaps some few frequency ranges Φ_m. In the case of the updating in a single frequency range Φ_m therefore the following is applicable

$$\delta H(z)[k] = (\gamma_m[k] - \gamma_m[\kappa_m]) \cdot H_m(z),$$

wherein κ_m designates the sampling interval in which the frequency range Φ_m has been updated the last time. Therefore in the predefined regular sampling intervals or, respectively, time intervals, preferably with the sampling rate of the input signal, not all, but solely selected coefficients are adapted, preferably exactly a single one.

For the selection of the frequency range or frequency ranges Φ_m to be updated at a certain sampling interval, in principle various possibilities exist. It is possible, e.g., to update respectively that frequency range Φ_m, for which $|c_m[k] - c_m[\kappa_m]|$ is at a maximum, or those frequency ranges Φ_m, in which these values exceed a certain threshold value, e.g., 1 dB. Another different possibility consists in the method that m simply time and again systematically or pseudo-randomly runs through all values from 1 to M.

In a preferred embodiment of the invention, by means of the correction values $r_1..r_n$ the following facts are taken into consideration: The noise suppression establishes attenuation values, which are only dependent on the modulation depths, not, however, on the signal levels themselves, as is correct for persons with a normal hearing. Persons with an impaired hearing, whose subjective perception of loudness, however, in general increases in a non-linear manner with the signal level, as a result will perceive a signal attenuation by a fixed value a_m differently distinct, depending on the signal level. In a serial processing, therefore in the case of a noise suppression with an immediately following compression amplification, this effect would be automatically corrected. Because here, however a parallel processing is taking place, the correction values $r_1..r_n$ are transmitted from the noise suppression to the compression amplification, in order to implement this correction. Thus in the signal

analysis for the noise suppression, attenuation-conditioned correction values r_n are determined for the N signal levels of the compression amplification and the calculation of the amplification values takes place with signal levels, which are reduced by these correction values. Thus, the compression amplification is corrected in accordance with the noise suppression. With this it is achieved that the signals optimally processed, by means of the noise suppression, for the person of normal hearing are individually correctly reproduced in the hearing range of each and every person with an impaired hearing.

This specifically signifies, that for every frequency range Φ_m in addition to the already available signal power s[k] also a as a result of the frequency-specific noise suppression reduced signal power u[k] is calculated. For the frequency ranges Φ_m contained in a frequency range F_n, the s[k] and the u[k] are separately added. From the logarithmic ratio of the two sums the valid logarithmic correction value r_n relative to F_n is obtained.

Figure 3 depicts a block diagram for a corresponding signal processing, as it takes place in the signal processing for the noise suppression 4 for determining the correction values r_n. A case is represented, in which three frequency ranges Φ_m of the noise suppression are contained in a frequency range of the compression amplification. In a block 31, in a known manner a signal power s[k] on the signal path 38 is determined and from it in block 32 a signal level, and from this in block 33 a modulation depth d_m and from this in Block 34 an attenuation value a_m. In block 35, the logarithmic attenuation value a_m is linearly scaled, and by multiplication with the signal power s[k] the reduced signal power u[k] on signal path 35 is calculated.

The reduced signal power u[k] is calculated for each one of the three frequency ranges, thus for y_m, y_{m+1}, y_{m+2} in parallel and added together in node 37. The signal powers s[k] of the three frequency ranges are added together in the summation point

39. The totals are logarithmically scaled in the blocks 40, respectively, 41 and in the subtraction 42 the correction value r_n is formed as a difference.

The device according to the invention preferably is at least partially implemented as an analogue circuit or based on a micro-processor or implemented with the utilisation of application-specific integrated circuits or with a combination of these techniques.

LIST OF DESIGNATIONS

1	First filter unit
2	Second filter unit
3	Signal processing for the compression amplification
4	Signal processing for the noise suppression
5	Combination unit
6	Controllable filter
7	Means for determining a compression amplification
8	Means for determining a noise suppression
X	Input signal
Y	Output signal
21	Power formation
22	Level calculation, logarithmic scaling
23	Subtraction
24, 24', 24"	Amplification function
31	Power formation
32, 40, 41	Level calculation, logarithmic scaling
33	Determination of modulation depth
34	Determination of attenuation value
35	Linear scaling
36	Reduces signal power u[k]
37, 39	Summation
38	Signal power s[k]
42	Subtraction